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EXAMINER	
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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No.	Applicant(s)	
	10/776,894	MINHAZUDDIN, MUNEYB	
	Examiner	Art Unit	
	ANDREW LAI	2616	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE *3* MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *24 April 2008*.

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-41* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-41* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application

6) ☐ Other: _____.

DETAILED ACTION

Claim Rejections - 35 USC § 103

1. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

2. Claims 1-5,7-9,12,14-15,17-27,29-31,34,36-37,39-42 are rejected under 35 U.S.C. 103(a) as being unpatentable over Shaffer et al (US 7,023,839, Shaffer hereinafter) in view of Graham et al (US 6,097,722, Graham hereinafter) and further in view of Johnsson (US 2002/0006131).

Shaffer discloses "system and method for dynamic codec alteration" (col. 1 lines 1-2) in "telecommunication systems" (col. 1 lines 16-17 and fig. 1) comprising the following features:

- **<u>With respect to Independent claims 1 and 23</u>**

Regarding claims 1, *a method for performing call admission control* (refer to fig. 1 and see "the H.323 gateway 106 generally provides ... and performs call setup and clearing", col. 4 lines 41-44, see further, e.g. fig. 6 depicting "ARQ 602" for call Admission Request), *comprising: …*

Regarding claim 23, *a call admission controller* (refer to fig. 1 "gatekeeper 108 and "BWAS [bandwidth allocation server] 109"), *comprising:*

a processor (fig. 3 "control processor 302") *operable to …*

Regarding claims 1/23: *(a) determining/determine* for the system *at least one of (i) a bandwidth utilization level* (refer to fig. 1 and see “the BWAS [bandwidth allocation server] 109 monitors system bandwidth usage”*, and (ii) an available bandwidth level* (“the BWAS 109 calculates the remaining network bandwidth”, col. 6 line 21) *and one or more Quality of Service or QoS metrics* (“the BWAS 109 saves the requested QoS levels for existing calls as well as the actual QoS level being provided”, col. 9 lines 28-30);

(b) comparing/compare the determined at least one bandwidth level to one or more selected thresholds (generally see "BWAS 109 can measure and track the network traffic to make the determinations of the relevant thresholds being crossed", col. 5 lines 23-25, and particularly see "compares the system bandwidth usage against the threshold value X", col. 5 line 67 - col. 6 line 1, and "[check] if the threshold X were to be exceeded such that 1 Mbps network bandwidth is remaining", col. 6 lines 34-35. It should be noted that Shaffer also discloses further that “if one or more new calls require a higher QoS, then the BWAS 109 determines whether lower QoS calls may be reset", col. 5 lines 30-33. This suggest that QoS may also be taken into account when performing call admission) *to determine whether a new live voice communication* ("performs call setup and clearing on both the LAN side and switched circuit network (e.g., public switched telephone network or PSTN) side", col. 4 lines 43-45, noting "PSTN" deals with *live voice communications*) *may be set up with a first selected codec* ("the BWAS 109 monitors system bandwidth usage and directs each H.323 terminal to

adopt a particular codec or coding algorithm according to bandwidth availability", col. 3 lines 5-8);

(3) when a new live voice communication may not be set up with the first selected codec (fig. 8 step 806 "BW [bandwidth] Avail ?" and associated "No" branch), *performing at least one of the following steps:*

(i) selecting/select a second different codec from among a plurality of possible codecs for the new live coice communication, wherein the second codec has a lower bit rate than the first codec;

(ii) changing/change an existing live voice communication from the first codec to the second codec; and

(iii) redirecting/reditect the new live communication from the first path to a second different path, wherein the second path does not include the first link

(fig. 8, continuing along above cited "No" branch, at step 812 where it is checked "if there exist connections whose QoS is presently more than needed or requested", col. 9 lines 44-46, and if "No", step 816 "make call with lower codec speed"; or if "Yes", step 814 "re-negotiate codec speed" for *existing communications* as further explained "If, however, the existing connections may be downgraded, the renegotiate lower codec speed process is undertaken in a step 814, ... and the call is made (step 808)", col. 9 lines 51-54).

Shaffer does not expressly disclose, regarding claims 1/23, that the step (a) determining/determine one of the bandwidth utilization level, available bandwidth level, and QoS metrics is with respect to *a* particular *path including a* particular *link.*

Graham discloses “bandwidth management processes and systems for asynchronous transfer mode networks using variable virtual paths” (col. 1 lines 1-4) wherein call “connection admission control” (col. 2 line 36) is implemented, comprising determining available bandwidth level with respect to *a particular path including a particular link* ("when a virtual channel connection is requested, it must be placed in a virtual path, so that the CAS software can determine if there is enough bandwidth Remaining in the virtual path to support the new virtual channel connection”, col. 2 lines 36-40).

It would have been obvious to one of ordinary skill in the art at the time of the invention to modify the method/system of Shaffer by adding the path specific bandwidth determining mechanism of Graham to Shaffer in order to provide more efficient system “which allows for greater use of the available capacity of networks, and particularly, transmission facilities with a network” (Graham, col. 4 lines 24-26).

Although Shaffer has disclosed considering comparing bandwidth limitation in call admission to one or more thresholds and suggested considering also QoS, neither does Shaffer nor Graham expressly disclose, regarding claims 1 and 23, such comparison takes into account bandwidth information *and* (in combination with) *one or more Quality of Service or QoS metrics* for call admission.

Johnsson discloses “arrangement for establishing connections through a network" (Title) based on bandwidth, which “means the number of bits per second that can be transmitted by a user of the connection” ([0004] lines 5-7), and QoS "in terms of

end-to-end delay and delay variation, packet loss ratio, etc." ([0004] last two lines). Johnsson's disclosure comprises:

Regarding claims 1 and 23, *comparing available bandwidth level for the first path and one or more Quality of Service or QoS metrics* [to certain thresholds] ("resources have to be allocated in the network to assure that each individual end-to-end AAL2 connection, for each direction is assigned a certain amount of bandwidth <u>and</u> that a certain level of quality of service (QoS) can be guaranteed for that connection", [0004] lines 1-5, emphasis added).

It would have been obvious to one of ordinary skill in the art at the time of the invention to further modify the method/system of Shaffer by adding bandwidth availability in combination with QoS factors of Johnsson in call admission in order to provide a refined call control mechanism which guarantees "that appropriate resources are available to ensure that the network can guarantee the bandwidth <u>and</u> the QoS associated with the connection" (Johnsson, [0005] lines 2-5, emphasis added).

- **<u>With respect to Dependent claims</u>**

Shaffer discloses the following features:

Regarding claims 2/24, *(c) (i) is performed and further comprising: receiving a request to place the live voice communication* (fig. 6 step 602 "ARQ" or "Admission Request");

setting up the live voice communication with the second codec (fig. 8 step 816 "make call with lower codec speed").

Regarding claims 3/25, *wherein each of a plurality of codecs has a corresponding bit rate and/or required bandwidth level* ("although the G.711 codec is the mandatory audio codec for an H.323 terminal, other audio codecs, such as G.728, G.729, G.723.1, G.722, MPEG1 audio, etc. may also be used", col. 3 lines 52-55, which codes are well known in the art to have different *bit rate* which in tern *require corresponding bandwidth level*) *and the selecting step comprises: comparing at least one of the available bandwidth level and the bandwidth utilization level with the plurality of bit rates and/or bandwidth level* ("if one or more new calls require a higher QoS (i.e., bandwidth), then the BWAS 109 determines whether lower QoS calls may be reset to still lower QoS codec", col. 5 lines 30-33); *and*

selecting the highest quality codec having a corresponding bit rate and/or bandwidth level permitted by the at least one of the available bandwidth level and the bandwidth utilization level ("the BWAS 109 may send an RAS command or H.245 signaling to the H.323 terminals to step down to the next fastest coding algorithm", col. 7 lines 45-47).

Regarding claims 4/26, *wherein the comparing comprises:*

comparing at least one of (i) a bandwidth utilization level and (ii) an available bandwidth level with one or more selected thresholds; and comparing one or more Quality of Service or QoS metrics with one or more selected thresholds (see discussion above regarding claims 1/23), *wherein the second codec has a bandwidth usage characteristic sufficient to satisfy the comparing steps* ("the BWAS 109 may send an RAS command or H.245 signaling to the H.323 terminals to step down to the next

fastest coding algorithm", col. 7 lines 45-47, noting that to be able to identify "the next fastest" codec, the codecs have to have *bandwidth usage characteristic sufficient* for *the comparing steps*).

Regarding claims 5/27, *wherein the comparing operation comprises:*

estimating an impact on the one or more QoS metrics from placing the new live voice communication with the second codec (refer to fig. 8 showing a series of steps when admitting a new call wherein step 802 shows "receive new QoS request level" determined primarily by the default codec to be used by the new call and then step 804 shows "compare requested QoS with available bandwidth", which step in combination with a later step 812 determining if existing "calls available" to use lowered codecs to meet "requested QoS with available bandwidth" will have to involve *estimating impact on QoS* due to the fact of *placing the new call with the second codec*, or "lower codec speed" shown in step 816 therein).

Regarding claims 7/29, *wherein substep (c)(ii) is performed* ("if the difference between the QoS levels meets a threshold, then the existing call(s) will have its (or their) codecs renegotiated to a lower level", col. 8 lines 65-57).

Regarding claims 8/30, *wherein, when the existing live voice communication was set up, the first and second codecs were identified as being acceptable to both endpoints.*

Regarding claims 9/31, *wherein substep (c)(ii) comprises: renegotiating with destination the codec to be used in the live voice communication.*

("the BWAS ... monitors bandwidth usage, and if there is a disparity between the bandwidth allocated to new connections versus ongoing ones ... the BWAS sends a lower codec speed message to all active H.323 entities. This causes the H.323 entities to renegotiate their codecs. The original calling party then selects a lower Speed codec and sends a message to the called party to proceed with H.323 codec negotiation", col. 2 lines 9-17. It should be noted that the fact both parties were in "ongoing" call indicates that a first codec was accepted by both, and the "renegotiation" indicates that a second (and lower speed) codec is also accepted by both).

Regarding claims 12/34, *wherein in the determining operation the bandwidth utilization level is determined* (refer to figs. 1 and 3 and see "the BWAS 109, in particular the bandwidth monitor 306, proceeds to monitor system bandwidth usage", col. 5 lines 62-64).

Regarding claim 14/36, *wherein the bandwidth utilization level is the end-to-end bandwidth* (since Shaffer discloses *end-to-end* codec negotiations, i.e. between calling and called parties, based on *bandwidth utilization level* monitoring, said *bandwidth utilization level* will also have to be *end-to-end*).

Regarding claims 15/37, *wherein in the determining step the available bandwidth level is determined* ("The BWAS 109 calculates the remaining network bandwidth", col. 6 lines 20-21).

Regarding claims 17/39, *wherein the available bandwidth level is the end-to-end bandwidth* ((since Shaffer discloses *end-to-end* codec negotiations, i.e. between

calling and called parties, based on *available bandwidth level* monitoring, said *available bandwidth level* will also have to be *end-to-end*).

Regarding claims 18/40, *wherein one or more QoS metrics is determined* (fig. 8 step 800 “receive QoS levels”).

Regarding claim 20/42, *wherein the available bandwidth level is the bandwidth allocable to live voice communications less the bandwidth utilization level* (this a obvious, and in fact intuitive, to one skilled in the art, because the term *available bandwidth* literally indicates remaining bandwidth that is not being used, in other words, *allocable bandwidth less the bandwidth utilization level*).

Regarding claim 21, *a computer readable medium* (fig. 3 “memory 308”) *operable to perform the steps of claim 1* (“The control processor 302 is couple to a memory 308 which is used to store bandwidth threshold information”, col. 5 lines 8-10, noting that said "threshold information" is used to *perform the steps of claim 1* as already discussed above with respect to claim 1).

Regarding claim 22, *a logic circuit* (fig. 1 “gatekeeper 108" and "BWAS 109”) *operable to perform the steps of claim 1* (see discussion above regarding claim 1).

Shaffer does not expressly but Graham does disclose:

Regarding claims 18/40, QoS is determined *for the first path*, or in other words *path*-specifically determined (refer to fig. 1A and see “virtual paths are grouped or pooled together for Clients A and B by a number of factors, such as Quality of Service (QoS) and bandwidth", col. 5 lines 29-31).

Johnsson does not expressly but Johson does disclose:

Regarding claims 19/41, *wherein one or more QoS metrics is at least one of packet delay, jitter, packet loss, the availability of Differential Services Code Point and RSCP status* ("QoS can be expressed in terms of end-to-end delay and delay variations, packet loss ratio, etc.", [0004] last two lines).

3. Claims 6/28, 10/32 and 11/33 are rejected under 35 U.S.C. 103(a) as being unpatentable over Shaffer in view of Graham and Johnsson as applied to claims 1, 2, 23 and 24 above, and further in view of Ho (US 6,452,922).

Shaffer in view of Graham and Johnsson discloses claimed limitations in section 2 above wherein Shaffer further discloses:

Regarding claims 6/28, *wherein, when there is no codec from among the plurality of codecs that satisfies the one or more thresholds, performing one or more of blocking the new live voice communication* (refer to fig. 6 and see "a calling H.323 terminal issues an Admission Request (ARQ) message to the gate keeper 108. In a step 604, the gatekeeper 108 accepts by issuing an Admission Confirm (ACF) message (it is noted that the gate keeper 108 could reject by responding with an Admission Reject (ARJ) message", col. 8 lines 18-22, noting that since Shaffer also disclosed lowering codecs as discussed above, it would only be obvious, and in fact intuitive, to use ARJ for *blocking the new call when there is no codec from among the plurality of codecs* can be further lowered down to).

Shaffer in view of Graham and Johnsson does not expressly disclose the other features of claims 6/28, and the features of claims 10/32 and 11/32.

Ho discloses that "an apparatus causes alternate connection of a telephone call directed an IP network” (Abstract lines 1-2) employing separate "IP interface" card and "PSTN interface” card (fig. 1 items 104 and 102). Ho’s invention comprises:

Regarding claims 6/28, *redirecting the new live voice communication from a packet-switched network to a circuit-switched network* (refer to fig. 2 and see "in this embodiment, the IP network interface 204 sends an ISDM call reject signal to the call processor 100 if the call cannot be connected through the IP network 108 because of a low QoS. The call processor 100 will then use an alternate connection through another network interface 202 according to ISDN protocol”, col. 2 lines 64-col. 3 line 2, noting that in fig. 2 explicitly shows that “interface 202” is "PSTN interface" connected to "public switched telephone network 106", well-known being *circuit switched network*).

Regarding claims 10/32, *wherein substep (c)(iii) is performed* [note: substep (c)(iii) recites *redirecting/redirect the new live voice communication from the first path to a second different path wherein the second path does not include the first link*] (refer to fig. 1 and see “a network interface card 104 … will cause a call directed to the card to be redirected to a different network 106 if the QoS for the call will be below the desired threshold”, col. 2 lines 41-45).

Regarding claims 11/33, *wherein the first link corresponding to a first set of port numbers and the second link to a second set of port numbers, wherein the first and second sets of port numbers are non-overlapping, wherein packets addressed to one of the first set of port numbers are directed along the first link and packets addressed to one of the second set of port numbers are directed along the second link* (refer to fig. 1

wherein "call processor 100" comprises two separate, thus *non-overlapping*, ports communicating via two separate, thus *non-overlapping*, *links* with two separate, thus *non-overlapping*, network "interface" cards, i.e. “IP” and "PSTN”. It is obvious as well as intuitive to one skilled in the art to appreciated for said separate or *non-overlapping ports* to have separate, thus *non-overlapping*, *first and second sets of non-overlapping port numbers*) *and wherein the redirection step comprises:*

selecting/select for the packetized live voice communication a port address (note that it is well known in the art that each port in a multi-ports communication unit, such as the “call processor 100” cited above, is identified by *a port address* or certain type of port ID) *from the first set of port numbers when a new live voice communication can be set up with the first selected codec and*

selecting/select for the packetized live voice communication a port address from the second set of port numbers when a new live voice communication cannot be set up with the first selected codec.

(still refer to fig. 1 and see "If the QoS that the IP network 1 108 has provided recently for test packets to be destination of the call to be routed is less than the desired QoS threshold, then the call is returned to the call processor 100 to be connected through another network 106 [noting that call to networks 108 and 106 uses above cited separate *ports, interfaces and links*]. Otherwise, the call is connected through the IP network 108).

It would have been obvious to one of ordinary skill in the art at the time of the invention to further modify the method/system of Shaffer by adding the call redirection

mechanism of Ho to Shaffer in order to provide more robust system capable of "monitoring the quality of service (QoS) of the IP network and connecting a telephone call over an alternate network, on a call by call basis" (Ho, col. 1 lines 60-62) which would offer an important improvement to overcome the "disadvantage of VoIP Networks" having the "variability of the quality of the signal received at the destination as determined by changing network conditions" (Ho, col. 1 lines 42-44).

4. Claims 13/35 and 16/38 are rejected under 35 U.S.C. 103(a) as being unpatentable over Shaffer in view of Graham and Johson as applied to claims 1 and 23 above, and further in view of Lachman, III et al (US 2002/0166063, Lachman hereinafter).

Shaffer in view of Graham and Johnsson discloses claimed limitations in section 2 above.

Lachman discloses "system and method for anti-network terrorism" (page 1 left column lines 1-2) using "a graph generated by Multi-Router Traffic Grapher (MRTG)" ([0140] lines 3-4).

Shaffer does not but Lachman does disclose:

Regarding claim 13/35, *wherein the bandwidth utilization level is determined by polling a local edge router.*

Regarding claims 16/38, *wherein the available bandwidth level is determined by polling a local edge router.*

(refer to fig. 17, which "illustrates a man screen GUI for a central monitoring station", [0038] lines 1-2, and see "through block 1704, the MRTG can poll the Router's

SNMP data and can chart the relative inbound/outbound bandwidth utilization", [0156] lines 4-6. It should be noted that knowing *bandwidth utilization* is equivalent to knowing *available bandwidth* because the latter is merely total bandwidth less the former).

It would have been obvious to one of ordinary skill in the art at the time of the invention to further modify Shaffer by adding the particular bandwidth monitoring method of Lachman to Shaffer in order to provide a better protected system "that can monitor incoming data packets fro a number of routers on a host network and that can detect a flood attack on an of the routers" (Lachman, [0014] lines 16-18).

Response to Arguments

5. Applicant's arguments with respect to claims 1 and 23 have been considered but are moot in view of the new ground(s) of rejection.

Applicant amended claims 1 and 23 in such a way that, when performing call admission, consideration is given to QoS metrics and bandwidth <u>all together</u> while they were claimed to be considered separately. Applicant then argues against Shaffer in view of Graham (Remarks page 8 seventh paragraph), "Neither Shaffer nor Graham teach suggest or disclose the use of the <u>combination</u> of both the QoS metrics and bandwidth level as claimed" (emphasis added).

This argument is moot in view of newly applied art of Johnsson wherein explicitly taught is comprised of "resources have to be allocated in the network to assure that each individual end-to-end AAL2 connection, for each direction is assigned a certain amount of bandwidth <u>and</u> that a certain level of quality of service (QoS) can be guaranteed for that connection" ([0004] lines 1-5, emphasis added).

Examiner also would like to point out that even in Shaffer, QoS consideration for call admission is also expressly provided. It is just that Shaffer does not explicitly say that QoS is taken into account in combination with bandwidth limitation. Such combination however would only have be obvious to one skilled in the art at the time of the invention, and Johnsson readily provided such express teaching and thus the argument is rendered moot.

6. Applicant's arguments with respect to all dependent claims have been considered but are moot in view of the new ground(s) of rejection.

Applicant argues (Remarks page 8 eighth paragraph), “The dependent claims are further distinguishable for at least the above reasons [those argued over for claims 1/23] and the additional feature(s) recited therein. For example:..."

As to “the above reasons”, see discussion in section 5 above, which renders the argument moot therein and thus moot for all dependent claims.

As to “additional feature(s)”, Applicant merely repeated the list of claims 2 through 20 without mentioning anything why those features are distinguishable from the applied arts. Repeating merely the list of claims without presenting reasons showing which element(s) of applied arts fail(s) to teach the claimed features essentially presents no arguments, and thus is moot. However, Examiner still went through each and every claim in view of previously applied arts, and comes to the conclusion that all claimed features have been disclosed in the various arts, singularly or in combination.

Conclusion

7. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure.

US 6,240,066 provides dynamic bandwidth and buffer management for multi-service ATM switches wherein bandwidth level and QoS are all considered together for connection admission.

8. Applicant's amendment necessitated the new ground(s) of rejection presented in this Office action. Accordingly, **THIS ACTION IS MADE FINAL**. See MPEP § 706.07(a). Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the date of this final action.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to ANDREW LAI whose telephone number is (571)272-9741. The examiner can normally be reached on M-F 7:30-5:00 EST, Off alternative Fridays.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Kwang Yao can be reached on 571-272-3182. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

/Andrew Lai/
Examiner, Art Unit 2616

/Kwang B. Yao/
Supervisory Patent Examiner, Art Unit 2616